Audio Quality Research

Outputs

- Technical publications documenting new research results.
- Subjective measurements and objective estimates of speech and audio quality.
- Algorithms and software for speech and audio coding and quality assessment.

Digital coding and transmission of speech and audio signals are enabling technologies behind many innovations in telecommunications and broadcasting including digital cellular telephone services, voice over Internet protocol (VoIP) services, and digital audio broadcasting systems. Speech signals can be coded and transmitted at rates as low as 4 kbit/s with good resulting quality. More general audio signals that include music and other sounds can be coded and transmitted with remarkably high fidelity at rates between 16 and 256 kbit/s per channel. In addition, coded speech and audio signals can be transmitted as data packets, thus sharing channel capacity (e.g., radio spectrum or wired network bandwidth) with other data streams and hence with other users.

In digital coding and transmission, one generally must trade off quality, bit-rate, delay, and complexity. In addition, the robustness of digital coding and transmission algorithms is critical in applications that use lossy channels. Important examples of lossy channels include those provided by wireless systems and those provided by the Internet. The ITS Audio Quality Research Program seeks to identify and develop new approaches that increase quality and robustness or lower bit-rate, delay, or complexity of digital speech and audio coding and transmission. The ultimate result of such progress is better sounding, more reliable, more efficient telecommunications and broadcasting services at lower costs.

In most digital speech and audio coding and transmission systems, a set of complex time-varying interactions among signal content, source coding, channel coding, and channel conditions make it difficult to define or measure speech or audio quality. The Audio Quality Research Program operates a subjective testing facility and runs controlled experiments to gather subjects' opinions of the speech or audio quality of various coding and transmission systems. Subjects provide their opinions through electronic forms, an example of which appears in Figure 1 above.

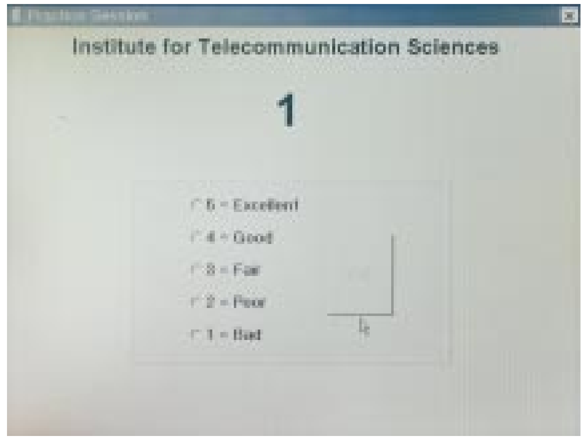

Figure 1. Example electronic response form used in subjective testing.

The Program has developed, verified, and patented tools for the objective estimation of telephone bandwidth speech quality. An example screen from one ITS-developed software tool is shown in Figure 2.

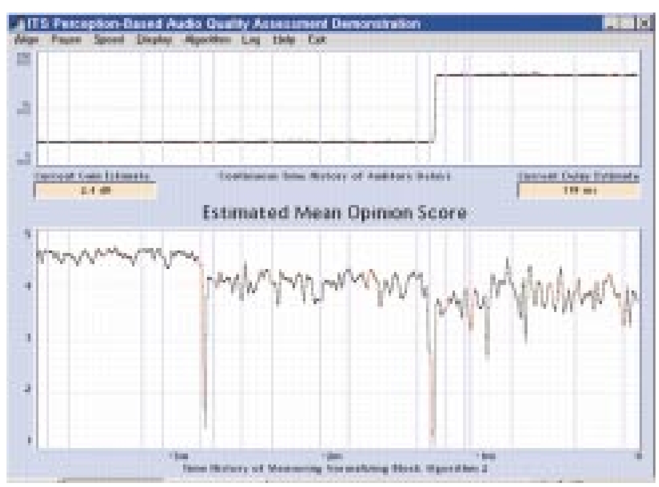

Figure 2. Example screen from ITS-developed software tool for objective estimation of perceived speech quality.

Throughout FY 2002, Audio Quality Research Program staff continued to apply and support such tools. Staff applied software tools, equipment and expertise in experiments aimed at finding relationships between the quality of speech delivered by VoIP systems and underlying network conditions.

Program staff also investigated several potential new innovations in speech and audio coding. In one investigation, the concepts of linear algebra were applied to time-frequency representations in order to generate novel decompositions of audio signals. These new decompositions display some desirable quantization properties and may ultimately lead to more efficient speech or audio coding schemes. Figure 3 uses 19 different colors to show how each of the 1024 different signal components in the decomposition can be linked to one of 19 fixed quantizers. The result is nearly optimal quantization of all 1024 signal components.

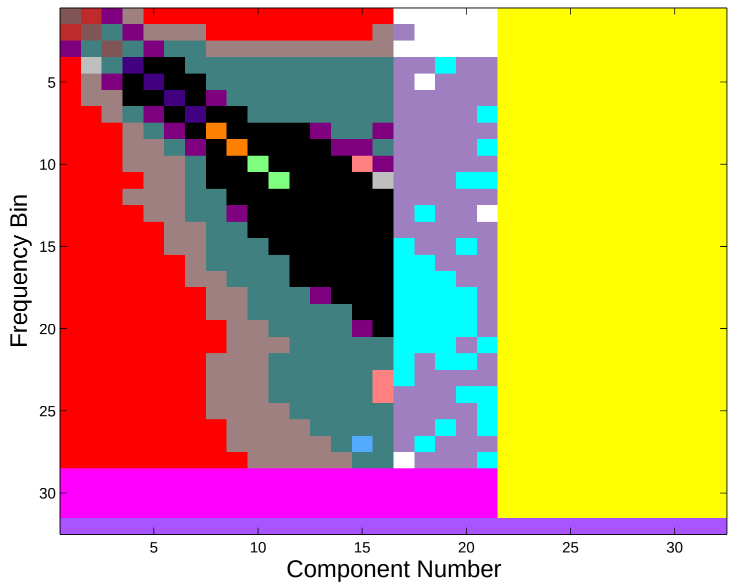

Figure 3. Example quantizer map for coding technique under study: audio signal has been decomposed into 1024 components, each component has been assigned to one of 19 quantizers indicated by one of 19 different colors.

The Audio Quality Research Program recently responded to a specific inquiry regarding the use of speech and audio coding algorithms for signals other than those they were designed for. For example, if a speech coding algorithm is given the sounds of laughter, or the sounds of a train, how badly will it distort each of these sounds? To answer the broader question in a concrete way, Program staff generated over 700 sound files and a set of tables linking to those files. Through these tables, a user can select example sounds and coders and hear first hand what a typical result might sound like.

Also in FY 2002, the Audio Quality Research Program staff performed significant upgrades to the ITS Audio-Visual Laboratories. An ad-hoc network of digital interconnections was replaced with a single unified digital audio infrastructure based on a 16 by 16 digital audio routing switcher. The Audio Quality Research Program continued to transfer technology to industry, Government, and academia throughout FY 2002. Program staff prepared publications, delivered invited lectures and presentations, provided laboratory demonstrations, and completed peer reviews for journals and workshops. More detailed Program results are available at
http://www.its.bldrdoc.gov/home/programs/audio/audio.htm

Recent Publications:

S.D. Voran, "Estimation of system gain and bias using noisy observations with known noise power ratio," NTIA Report 02-395, Sep. 2002.

S.D. Voran, "An iterated nested least-squares algorithm for fitting multiple data sets," NTIA Technical Memorandum TR-03-397, Oct. 2002.

S.D. Voran, "Compensating for system gain: Motivations, derivations, and relations for three common solutions," NTIA Technical Memorandum TR-03-398, Oct. 2002.

S.D. Voran, "Channel-optimized multiple-description scalar quantizers for audio coding," to appear in *Proc. IEEE 10th Digital Signal Processing Workshop*, Pine Mountain, GA, Oct. 2002.

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